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NEW PROVISIONAL APPLICATION TRANSMITTAL LETTER

Sir:

Transmitted herewith for filing is the Provisional Patent Application of Inventor(s):

	Marcos C. Tzannes
Residence:	121 LaEspiral Orinda, CA 94563
Citizenship	U.S.A.
Post Office Address:	Same as above

PATENTS
T3653-8993PV01

For: On-Line Impulse Noise Protection (INP) Adaptation

Enclosed are the following papers required to obtain a filing date under 37 C.F.R. §1.53(c):

__ Sheets of Informal Drawings
10 Pages of Specification, Drawings & Tables
__ Claims
1 Appendices

The following papers, if indicated by an ☒, are also enclosed:

☐ A Declaration and Power of Attorney
☐ An Assignment of the invention
☐ An Information-Disclosure Statement, Form PTO-1449 and a copy of each cited reference
☐ A Small-Entity Declaration
☒ A Certificate of Express Mailing, Express Mail Label No. ET 104264368 US

Basic Fee: $160

☒ A check in the amount of $160 is enclosed to cover the Filing Fee.

Please address all communications and telephone calls to the undersigned.

Respectfully submitted,

Gail Leonick

Gail Leonick
Aware, Inc.
40 Middlesex Turnpike
Bedford, Mass. 01730

PATENTS
T3653-8993PV01

Summary of Appendices

1. ITU – Telecommunication Standardization Sector
   Study Group 15
   Source: SBC, BellSouth, BT
   Title: Proposed revisions to G.992.4 and G.992.3&5 regarding impulse noise protection.

PATENT
T3653-8993PV01

UNITED STATES PROVISIONAL PATENT APPLICATION

Of

Marcos C. Tzannes

for a

On-Line Impulse Noise Protection (INP) Adaptation

On-Line Impulse Noise Protection (INP) Adaptation
By Marcos Tzannes

Background

Communication systems often operate in environments with Impulse Noise. Impulse Noise is a short-term burst of noise that is higher than the normal noise that typically exists in the communication channel. For example, DSL systems operate on telephone lines and experience Impulse Noise from many external sources including telephones, AM radio, HAM radio, other DSL services on the same line or in the same bundle, other equipment in the home, etc. It is standard practice for communications systems to use Interleaving in combination with Forward Error Correction (FEC) to correct the errors caused by the Impulse Noise.

One of the problems with these communications systems is that they use traditional Signal to Noise Ratio (SNR) measurement techniques to determine the SNR of the channel. These traditional techniques assume that the noise is stationary and does not contain non-stationary components such as Impulse Noise. The most common method for measuring SNR is to calculate the mean-squared error of the received signal based on a known transmitted signal (see the ADSL series of ITU G.992.x standards and the VDSL series of ITU G.993.x. standards). These traditional methods for measuring SNR do not correctly measure the impact of Impulse Noise and do not have the capability to determine how the system should be configured to handle this Impulse Noise.

As an example, ITU-T Contributions SS-049 (attached) has proposed that there is a need in ADSL and VDSL systems to provide robust, error free performance in the presence of high real-world impulse noise. SS-049 proposes that the standard *Impulse Noise Protection (INP)* values are extended to values of 4, 8, 16 and 32 in order handle these high levels of impulse noise. INP is defined in the ADSL2 Standard G.992.3, which is incorporated herein by reference in its entirety, as the number of Impulse Noise corrupted DMT symbols that can be corrected by the FEC and Interleaving configuration. Specifically G.992.3 defines the following variables:

INP=1/2*(S*D)*R/N

S=8*N/L

Latency (or delay)=S*D/4

Line Rate (in kbps)=L*4

where N is the Codeword Size in bytes, R is the number of parity (or redundancy) bytes in a codeword, D is the interleaver depth in number of codewords and L is the number of bits in a DMT symbol. If K is the number of information bytes in a codeword then N=K+R and the User Data Rate is approximately equal to L*4*K/N.

SS-049 proposed that the higher INP values are achieved by increasing the amount of FEC redundancy while keeping the same system latency (and same Interleaver Memory) at the expense of user data rate or excess margin. Since, on phone lines without excess margin, there is clearly a tradeoff between high INP values and user data rate, it is best to

try to maximize the user data rate by finding the minimum INP value that will provide adequate impulse noise protection. The approach that is currently proposed in SS-049 involves the steps of:

1. The operator (or service provider) configures the ADSL connection with a specific INP value
2. The ADSL connection is initialized and transceivers enter into while in steady-state data transmission (also known as Showtime for DSL systems)
3. If the connection is stable (error-free) then the ADSL service is acceptable and the process ends. If there are bit errors then the process is repeated by going back to step #1.

An exemplary problem with process is that it is time consuming and can result in sub-optimum user data rates. To illustrate this lets take the following examples:

Example #1 (Time consuming process): Assume that for a particular DSL connection there is high impulse noise and the required INP is 8. As a result, if the service provider uses a first INP configuration of 2, the DSL connection will not be error free. Therefore the service provider needs to configure a higher INP value and reinitialize the connection. If a value of 4 is used as the second INP value, it still will not provide adequate impulse noise protection and the bit errors will occur. So the service provider will once again need to configure a higher INP value and so on, until the correct value of 8 is configured. Obviously the connection needs to be re-initialized every time there is a new INP configuration and this trial and error technique is very time consuming.

Example #2 (Sub-optimum User Data Rates): Assume that for a particular DSL connection there is high impulse noise and the required INP is 4. As a result, if the service provider uses a first INP configuration of 2, the DSL connection will not be error free. Therefore the service provider needs to configure a higher INP value and reinitialize the connection. However, in order to save time and not go through a number of initializations as in Example #1, the service provider simply configures the system to the maximum INP value of 32. Obviously there will be no errors with INP=32 since this connection only needed an INP value of 4. But as a result the user data can be greatly degraded since the additional FEC redundancy will be 3 times higher than what is actually needed. For example, if the an INP of 4 requires 10% FEC redundancy, an INP of 32 requires 40% FEC redundancy which results in a 30% decrease user data rate.

Overview of an Exemplary Embodiment of the Invention

An exemplary embodiment of this invention broadly describes a new method and system with the capability to determine the impact of Impulse Noise on a communication system and the capability to determine how the system should be configured to handle this Impulse Noise.

According an exemplary embodiment of this invention, the impact of impulse noise is determined by transmitting and receiving using a plurality of different Forward Error Coding (FEC) and Interleaving Parameter settings. For each FEC and Interleaving Parameter (FIP) setting the received signal quality is determined by, for example,

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detecting if there are bit errors after the receiver performs the FEC decoding and deinterleaving. Based on this, the appropriate FIP setting is selected and used for transmission and reception.

As described above, DSL systems (such as the one defined in ADSL G.992.x or VDSL G.993.x) use FIP parameters denoted as N, K, R and D. For these systems an FIP setting can be characterized by the set of parameters (N, K, R, D). Using these parameters, the Burst Error Correction Capability (BECC) in bytes can be simply calculated as BECC=D*R/2 bytes where BECC is defined as the number of consecutive byte errors that can be corrected by the receiver. Note that if the receiver uses more intelligent decoding schemes (e.g. erasure detection) it is possible to correct even more than D*R/2 bytes. It also follows from above that INP=BECC/L. For example a setting of (N=255,K=239,R=16,D=64) would correspond to N=255 bytes, K=239 bytes, R=16 bytes, D=64 codewords and in this case the receiver can correct BECC=D*R/2=64*16/2=512 bytes. If L=256*8 bits, this corresponds to a line rate of approximately 4*(256*8)=8.192 Mbps for standard ADSL and VDSL systems and INP=512/256=2.

According exemplary embodiments of this invention, a plurality of FIP settings are used for transmission and reception. In an exemplary embodiment, the system transitions from one FIP setting to another FIP setting is completed without going through a new initialization procedure such as the one specified in traditional xDSL systems. For example, an xDSL system that implements the method of this invention could start using a FIP setting of (N=255,K=247,R=8,D=64) and then change to a setting of (N=255,K=239,R=16,D=64) without completing a new initialization procedure. Note that the first FIP setting has a BECC=256 bytes and the second setting has a BECC=512 bytes, which means, for example, that the second setting can correct an impulse that causes twice the as many byte errors as the first FIP setting. On the other hand, the first FIP setting has less FEC parity (overhead), which results in a higher information (net) data rate for the user. This can also be seen by the fact that K, the number of information bytes per codeword, is higher for the first FIP setting. For each of the FIP settings the receiver detects whether there are bit errors after the decoding and deinterleaving process. This detection can be done by, for example, performing a Cyclic Redundancy Check (CRC) after the decoding/deinterleaving process as is defined in ITU standard G.992.x. In general, a CRC is a well-known method for detecting bit errors. Since Impulse Noise occurs at random times, the system operates using a particular FIP setting for a period of time that is sufficient to encounter the Impulse Noise. In the simple example directly above, only the K and R values were modified. This invention is not limited in any way in this manner but in fact any combination of FIP parameters can be modified in the process.

The process of determining the impact of impulse noise by transmitting and receiving using a plurality of FIP settings can be done while in steady-state transmission (also known as Showtime for ADSL systems) when user information bits are being transmitted. Alternatively the process of determining the impact of impulse noise by transmitting and receiving using a plurality of FIP setting can be done during a special

Impulse Noise Training period during which the system is not actually transmitting user data. For example, in this special Impulse Noise Training period the system could transmit idle ATM cells or HDLC flags or a predefined pseudorandom bit sequence.

Throughout this invention the term "transmitter" has the same meaning as the term "transmitting modem". Throughout this invention the term "receiver" has the same meaning as the term "receiving modem".

Detailed Description: On-Line INP Adaptation

According to an exemplary embodiment of this invention, the communication system adapts the INP setting on-line by operating using a series of different FIP settings. For each FIP setting the system determines if the appropriate amount of impulse noise protection (INP) is being provided. Based on these determinations the system selects a particular FIP setting for regular operation.

This On-Line INP adaptation procedure comprises the following exemplary steps:

1. The DSL system completes regular initialization and starts transmitting and receiving using a first FIP setting. For DSL systems this FIP setting is selected by the receiver and is based on the min/max data rate, max latency and min INP values as configured by the service provider via the CO-MIB (see G.992.3). Proceed to step 2.
2. The system operates using this FIP setting for a period of time T1. Proceed to Step 3.
3. The receiver detects if bit errors have occurred using this FIP setting for Decoding and Deinterleaving during the period of time T1. For example the receiver could use a CRC to detect bit errors. If there are no bit errors then the current INP setting is adequate and there is no need to change the INP setting on-line and the On-Line INP adaptation procedure ends. On the other hand, if there are CRC errors proceed to step 4.
4. Since there are bit errors, the service provider or user may choose to request an on-line increase of the INP setting (for example double the INP setting). Based on this request the INP setting would be increased by on-line modification of the FIP parameters. Go to step 5.
5. In order initiate the change in the INP setting, the ATU-C may send a message to the ATU-R that specifies the new INP value. Go to step 6.
6. The receiver sends a message to the transmitter indicating the new FIP setting that satisfies the new INP setting. Go to step 7.
7. The transmitter and receiver transition to the new INP setting by starting to use the new FIP parameters for transmission and reception, respectively, at a synchronized point in time. This synchronization could be done in a number of different ways (see section below). Go to step 8.
8. The system operates using this new FIP setting for a period of time T2. Proceed to Step 9.
9. The receiver detects if bit errors have occurred using this FIP setting for Decoding and Deinterleaving during the period of time T2. If there are no bit errors detected by the receiver during the period of time T2 then the new INP setting is adequate and there is no need to change the INP setting on-line and the On-Line INP adaptation procedure ends. On the other hand, if there are bit errors go back to step 4 and repeat the process to select a new INP setting.

The On-line INP adaptation process could be repeated as many times as desired until an INP setting is selected that provides the required impulse noise immunity.

There are several important points regarding these exemplary embodiments:

- The transition between different FIP settings is done without reinitializing the transceivers using a lengthy initialization procedure such as is used in ADSL and VDSL systems.
- The transition between FIP settings can be synchronized between the transmitter and the receiver so that the receiver can determine when to use start FEC decoding using the new FIP settings for K and R. The transition could be synchronized using a number of methods. The section below describes some of these methods.
- Alternatively the transition could be done without synchronization in which case the receiver would have to determine when the new FIP setting are used by some other means, e.g. by FEC decoding using both FIP settings and determining which one is being used by calculating whether the codeword is correct with one setting or the other.
- These steps can be performed during regular steady state transmission (also known as SHOWTIME in ADSL) using actual user data or, for example, idle ATM cells.
- Alternatively these steps can be performed during a special Impulse Noise Training period during which the system is not actually transmitting user data. For example, in this special Impulse Noise Training period the system could transmit predefined pseudorandom bit stream or, for example, idle ATM cells or HDLC flags.
- The length of the time periods T1, T2, etc can be controlled by the receiver. The receiver could control the length of these time periods by for example sending a message that specifies how long the transmitter should transmit using a particular FIP setting. This length can be defined, for example, in terms of the number of DMT symbols or the number FEC codewords. For example, the message could indicate that 200 DMT symbols should be sent for all FIP settings or 300 FEC codewords should be sent for all FIP settings. Alternately the message could indicate a different number of DMT symbols or FEC codewords for each FIP setting.
- The length of the time periods T1, T2, etc can be controlled by the transmitter. The transmitter could control the length of these time periods by for example sending a message to the receiver that specifies how long the transmitter will transmit using a particular FIP setting. This length can be defined in terms of the number of DMT symbols or the number of FEC codewords. For example, the message could indicate that 200 DMT symbols will be sent for all FIP settings or 300 FEC codewords will be sent for all FIP settings. Alternately the message could indicate a different number of DMT symbols or FEC codewords for each FIP setting.
- The length of the time periods T1, T2 and T3, etc can be also controlled by the service provide or the user. For example the service provider could configure through the CO-MIB that a minimum of time of X=10 seconds be used for testing

each FIP setting. The service provider could use knowledge of the nature of the impulse noise (e.g. how often the impulse noise occurs) to determine these times.

- In Step #4 of the INP Adaptation process the service provider chooses to request a new INP setting. Alternatively the receiver can make this choice. In this case, the receiver could continually adapt the INP value as described above until the are no bit errors due to impulse noise.

Synchronizing the modification of the FEC and interleaving parameters

As stated in exemplary embodiments above, the receiver and transmitter can synchronize the modification of the FEC and interleaving parameters so that they both start using them at the same instant in time. There are several ways to do this.

Synchronizing using FEC codeword counters: In this example, the receiver and transmitter synchronize the change by counting FEC codewords from the beginning of Showtime and the transition occurs on a specific FEC codeword counter value that is known by both the transmitter and the receiver. Prior to this the receiver or the transmitter will send a message to the other side indicating the FEC codeword count value on which the FIP parameters will be modified. For example the exemplary process is as follows:

1. The transmitting modem enters Showtime and starts a counter that counts the number of transmitted FEC codewords, where the first codeword transmitted has a count value of 0, the second transmitted codeword has count value of 1 and so on. From practical implementations this counter will probably have a finite length, for example 0 to 1023 (10 bits) so that when the value of 1023 is reached, on the next FEC codeword the counter starts at the value of 0 again.
2. Likewise, the receiving modem enters Showtime and starts a counter that counts the number of received FEC codewords, where the first codeword received has a count value of 0, the second received codeword has count value of 1 and so on. From practical implementations this counter will probably have a finite length, for example 0 to 1023 (10 bits) so that when the value of 1023 is reached, on the next FEC codeword the counter starts at the value of 0 again.
3. At some point in time it is determined that a new a FIP setting is needed due to the presence of Impulse Noise on the line. This determination can be done by the receiving modem, the transmitting modem, the user or service provide (operator).
4. The receiving modem sends a message to the transmitting modem indicating the new FIP setting to be used for transmission and reception. Alternatively, the transmitting modem sends a message to the receiving modem indicating the new FIP setting to be used for transmission and reception
5. The receiving modem sends a message to the transmitting modem indicating the FEC codeword counter value on which the new FIP settings are to be used for transmission and reception. Alternatively, the transmitting modem sends a message to the receiving modem indicating the FEC codeword counter value on

which the new FIP settings are to be used for transmission and reception. For example the message could indicate that when the codeword counter equals 501 the new FIP setting will be used for transmission and reception.

6. When the transmitter FEC codeword counter equals the value indicated in the message, the transmitting modem uses the new FIP settings for transmission.
7. Likewise, when the receiver FEC codeword counter equals the value indicated in the message, the receiving modem uses the new FIP settings for reception.

Synchronizing using a Flag Signal: In this example, the receiver and transmitter synchronize the change using a Flag or Marker signal (similar to the signal used in the ADSL2 G.992.3 OLR protocol.) This protocol may be more desirable that using an FEC codeword counter because it has more immunity to impulse noise. In this case the receiver and transmitter would start using the new FEC and Interleaving parameters on a predefined FEC codeword boundary following the SyncFlag. For example the process is as follows:

1. While transmitting using a first INP setting, it is determined that a new a FIP setting is needed due to the presence of Impulse Noise on the line. This determination can be done by the receiving modem, the transmitting modem, the user or service provide (operator).
2. The receiving modem sends a message to the transmitting modem indicating the new FIP setting to be used for transmission and reception. Alternatively, the transmitting modem sends a message to the receiving modem indicating the new FIP setting to be used for transmission and reception
3. The transmitting modem sends a Flag or Marker Signal to indicate that the new FIP setting are to be used on a predetermined number of DMT symbols following the transmission of the Flag or Marker Signal. For example the Flag signal could be an inverted Sync Symbol, or SyncFlag, as is used in the ADSL2 G.992.3 OLR protocol.
4. The transmitting modem start using the new FIP settings for transmission on the predetermined number of DMT symbols following the transmission of the Flag or Marker Signal.
5. Likewise, the receiving modem start using the new FIP settings for reception on the predetermined number of DMT symbols following the reception of the Flag or Marker Signal.

Example #1 of On-Line INP adaptation FIP Setting

This section describes an example of FIP settings for On-Line INP adaptation for DSL. In this example only the number of information bytes in a codeword (K) and the number of parity bytes in a codeword (R) are updated on line. The Codeword Size (N) and

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Interleaver Depth (D) are not changed. This means that the latency (and interleaver memory size) and the line rate are not modified on-line. Since N=K+R this places restrictions on the allowed values for K and R.

1st Setting – {Approximate User Data Rate= 3.968 Mbps, Line Rate=4.096 Mbps, N=128, K=124, R=4, S=1, D=64, Latency=16 msec, INP=1}

2nd Setting - {Approximate User Data Rate= 3.840 Mbps, Line Rate=4.096 Mbps, N=128, K=120, R=8, S=1, D=64, Latency=16 msec, INP=2}

3rd Setting - {Approximate User Data Rate= 3.584 Mbps, Line Rate=4.096 Mbps, N=128, K=112, R=16, S=1, D=64, Latency=16 msec, INP=4}

The On-line INP adaptation process is restricted to only modify the number of information bytes in a codeword (K) and the number of parity bytes in a codeword (R). The FEC Codeword Size (N=K+R) and Interleaver Depth (D) are not changed. This means that the latency (or interleaver memory size) and the line rate are not modified on-line. However the user data rate will change during the process since K is being modified. Also since the line rate and the FEC codeword size are not modified, the S value does not change in the process. It is important to note that with these constraints, the On-line INP adaptation process can be done in a seamless manner (no bit errors and service interruption). This means provided that the modification if the FIP setting is restricted to K and R, the transition between FIP settings can be done in a seamless manner. This is the case because if the codeword size N and the interleaver depth D are not modified, the transition can happen without the problem of "interleaving memory flushing." Interleaver memory flushing is a well-known problem in which errors occur because interleaver and deinterleaver memory locations are overwritten due to on-line changes in the codeword size (N) and or interleaver depth (D).

Example #2 of On-Line INP adaptation FIP Setting

This section describes an example of FIP settings for On-Line INP adaptation for DSL. In this example only the Codeword Size (N) and the number of parity bytes in a codeword (R) are updated on line. The number of information bytes in a codeword (K) and Interleaver Depth (D) are not changed and therefore the user data rate does not change. This means that the latency (and interleaver memory size) and the line rate are modified on-line. Since N=K+R this places restrictions on the allowed values for K and R.

1st Setting – {Approximate User Data Rate= 3.968Mbps, Line Rate=4.096 Mbps, N=128, K=124, R=4, S=1, D=64, Latency=16 msec, INP=1}

2nd Setting - {Approximate User Data Rate= 3.968 Mbps, Line Rate=4.224 Mbps, N=132, K=124, R=8, S=1, D=64, Latency=16 msec, INP=2}

3rd Setting - {Approximate User Data Rate= 3.968 Mbps, Line Rate=4.480 N=128, K=124, R=16, S=1, D=64, Latency=16 msec, INP=4}

In this example the Line Rate is modified on-line. For this reason it is necessary to also complete a Rate Adaptation process in order to complete this On-Line INP adaptation. A method for Seamless Rate Adaptation is described in US Patent 6,498,808 which is incorporated herein in its entirety.

While these examples restrict the changes to a subset of the FIP parameters, they can obviously be extended to cover any combination of the FIP parameters (N, K, R and D). For example, the value of D could also be modified in addition to the values of K, R and N. This could result in a change in the required interleaver memory and latency. In order to keep the memory and latency constant it is necessary to change the codeword size (N) accordingly when changing the interleaver depth (D). For example, if the interleaver depths is changed from D=64 to D=128, the Codeword size would have to be decreased by a factor of 2 so that overall latency is constant.

The above-described communication system can be implemented on wired or wireless telecommunications devices, such a modem, a multicarrier modem, a DSL modem, an ADSL modem, an XDSL modem, a VDSL modem, a multicarrier transceiver, a wired or a wireless wide/local area network system, or the like, or on a separate programmed general purpose computer having a communications device. Additionally, the systems, methods and protocols of this invention can be implemented on a special purpose computer, a programmed microprocessor or microcontroller and peripheral integrated circuit element(s), an ASIC or other integrated circuit, a digital signal processor, a hard-wired electronic or logic circuit such as discrete element circuit, a programmable logic device such as PLD, PLA, FPGA, PAL, modem, transmitter/receiver, or the like. In general, any device capable of implementing a state machine that is in turn capable of implementing the methodology illustrated herein can be used to implement the various communication methods, protocols and techniques according to this invention.

Furthermore, the disclosed methods may be readily implemented in software using object or object-oriented software development environments that provide portable source code that can be used on a variety of computer or workstation platforms. Alternatively, the disclosed system may be implemented partially or fully in hardware using standard logic circuits or VLSI design. Whether software or hardware is used to implement the systems in accordance with this invention is dependent on the speed and/or efficiency requirements of the system, the particular function, and the particular software or hardware systems or microprocessor or microcomputer systems being utilized. The communication systems, methods and protocols illustrated herein however can be readily implemented in hardware and/or software using any known or later developed systems or structures, devices and/or software by those of ordinary skill in the applicable art from the functional description provided herein and with a general basic knowledge of the computer and telecommunications arts.

Moreover, the disclosed methods may be readily implemented in software executed on programmed general purpose computer, a special purpose computer, a microprocessor, or the like. In these instances, the systems and methods of this invention can be implemented as program embedded on personal computer such as JAVA® or CGI script, as a resource residing on a server or graphics workstation, as a routine embedded in a dedicated communication system, or the like. The system can also be implemented by physically incorporating the system and method into a software and/or hardware system, such as the hardware and software systems of a communications transceiver.

ITU - Telecommunication Standardization Sector **SS-049**

STUDY GROUP **15**

Singapore, January 19-24 2004

Question: 4/15

SOURCE*: SBC, BellSouth, BT

TITLE: Proposed revisions to G.992.4 and G.992.3&5 regarding impulse noise protection

ABSTRACT

SBC field experience has shown that some ADSL lines fail to provide reliable service due to high level intermittent noise that has frequent re-occurrence. This differs from the classic concept of short duration and infrequent impulse noise. SBC field experiments have determined that many trouble cases due to intermittent noise can be effectively resolved by setting the ADSL coding parameters to provide a high level of coding redundancy. The necessary coding can reach 50% redundancy in some cases, while having an additional latency of 10 ms or less. The necessary coding redundancy exceeds the current specifications in the ITU ADSL Recommendations G.992.3 and G.992.5. It is proposed that the "minimum impulse noise protection (INP)" parameter in these Recommendations be revised to allow values up to 32 symbols and to change its name to "impulse strength indication".

1. Discussion

G.997.1 defines a parameter in Section 7.3.2.2 called "minimum impulse noise protection (INP)" that is measured in symbol periods and currently allowed to take one of the 4 values 0, .5, 1, and 2 symbols (corresponding to 0 125, 250, and 500 microseconds of a theoretical impulse). Section 7.3.2.3 also defines a maximum delay. The parameter is passed in G.994.1 and further described in Annex K of G.992.3. Both of these parameters are active in G.992.3/4/5 and actually exchanged during the "handshake" portion of training and so specified from modem to modem via the NPAR(3) octet of G.994.1. It is proposed that this section and parameter be renamed "impulse strength indication" and for the downstream direction takes on the additional values 4, 8, 16 and if possible values between 16 and 32. Note that there NPAR(3) uses bits 3

* Tom Starr Tel: +1 (847) 248-5467
SBC Email: tom.starr@ameritech.com

John Cioffi Tel: +1-925-823-2910
SBC Email: jc7513@sbc.com

Gary Tennyson Tel: +1-205-985-6087
BellSouth Email: gary.tennyson@bellsouth.com

Kevin Foster Tel: +44-1473-642-986
BT Email: kevin.t.foster@bt.com

and 4 today, allowing only the present 4 options. However, bit 5 is fortunately reserved by the ITU and so could be additionally allocated to INP also, thus allowing the desired 8 options proposed here. The ratio of this INP parameter to interleaving depth will then determine the codeword block size. The codeword block size will be supplied by the receiving modem back to the transmitter in a message known as C/R-PAREMS of the exchange of G.992.3 and G.992.5 ADSL modems. The specific parameters that will reflect the additional intermittent-noise protection will be the $B_{p,n}$ and $R_{p,n}$ that are determined by the receiver and passed to the transmitter in the usual manner.

A loading-algorithm engineer could use the following formulas to convert the supplied maximum delay and now renamed "impulse strength indication INP" into the usual FEC parameters (where $t = R$, the number of parity bytes chosen by the receiver, if erasures are used and $t = R/2$ if no erasures are used and rate includes information and synch/control bits)

$$D' = \text{depth in bytes} = \frac{(INP/4)\times\left(\frac{\text{rate in kbps}}{8}\right)}{\text{t}} \qquad (1)$$

$$N = \frac{(\text{delay in ms})\times\left(\frac{\text{rate in kbps}}{8}\right)}{(D'-1)} + 1 \approx \frac{\text{delay}}{\text{burst length}}\cdot t \qquad (2)$$

Then the parameters M, T, L, etc can be computed and S inferred for all latency paths as well as the values of B for each frame bearer and latency path. Also once N is known, then $D = ND'$. Even approximate adherence to the above mathematical rules should solve the problem. Service providers may well train the modem several times, each time increasing the INP value until zero or small numbers of code violations and/or error seconds are observed over a time period determined by the service provider.

We note here that even with the choice of smallest delay (4 ms) and largest burst length in the existing G.992.3 provides the relation

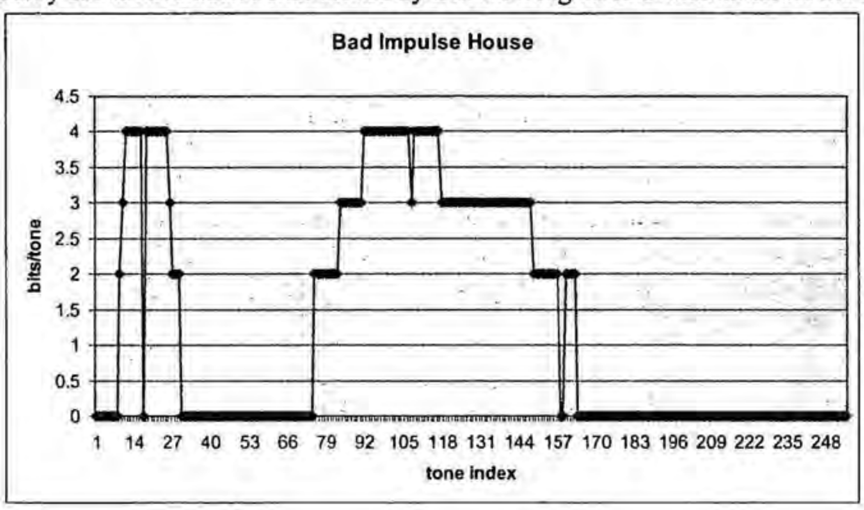
$$N \approx 8\cdot t$$

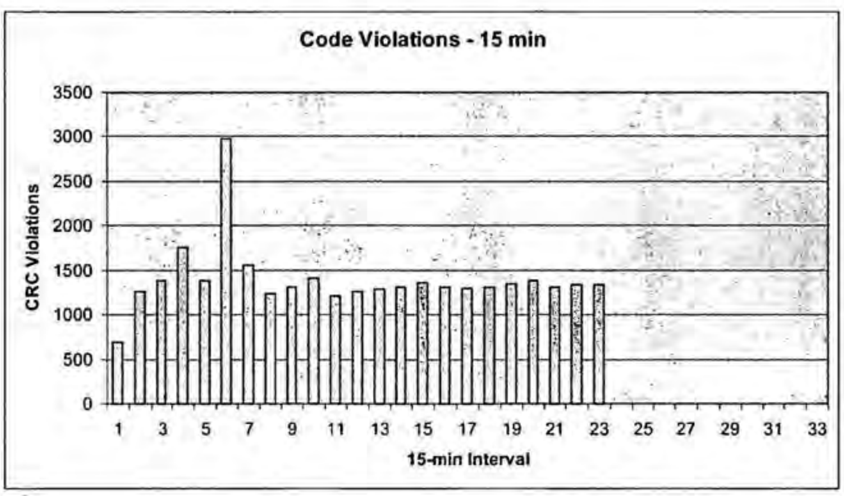
in equation (2) above. Since t is usually 16 in well-designed systems that have maximum parity then, the smallest codeword length is N=128, much larger than the values such as 32, 48, 64, and 80 that are effective for chronic lines with intermittent noise. The addition of the new values of INP=4,8,16, or 32 allows this expression to allow very small N values essentially providing service providers with the ability they need to correct a line undergoing difficult intermittent noise (even if at low data rate). The value of INP=8 allows codeword lengths as low as 32, and the higher values allow very low data rates on severely chronic lines.

An INP setting of 4 or larger will imply that impulse protection is more important than maximum data rate and so the equipment will connect at the highest data rate that maintains the impulse protection level indicated regardless of the net data rate. Service providers are expected to use the INP setting of 4 only for lines with severe noise, where the lower data rate is a better choice than not having reliable service.

2. Sample Example

The following example was taken from chronic lines reported by the SBC Broadband Tools. The customer operates in 768 kbps interleave profile, but has 28 dB margin downstream (and a maximum attainable data rate of over 4.5 Mbps reported). While the margin is high, the bit distribution clearly indicates that an intermittent noise is affecting frequencies below 400 kHz so often that the modem will not load into those frequencies. Even as such with the high margin indicated, the error seconds just below indicate a severe problem on the line. This line will operate with no errors at 1.5 Mbps if the for the combination $N=24$ and $R=16$. Such a setting is not a current possible selection by service providers if DSLAM profiles or MIBS do not have this choice (yet). The customer probably switched to cable-modem by the time this note was read, but easily could have been better served by DSL at a higher rate with no errors in ADSL1.





In specially investigating about 50 customers, the authors found 2 other lines that also could only be fixed (and would run at least 1.5 Mbps with zero errors instead of being out of service or chronic) by lowering the N parameter (or equivalently choosing a larger value for the "impulse strength" in ADSL2).

3. Proposal

1- Agree to an amendment for corrigendum to ITU G.992.3, G.992.5 G.994.1, G.997.1 to change the name of the parameter "minimum impulse noise protection (INP)" to "impulse strength indication".

2- In the above Recommendations, revise the definition of this parameter to have the additional values 4, 8, 16 and if possible values between 16 and 32.

3- In ITU G.994.1 NPAR(3) for this parameter, define bit 5 to allow for the additional defined values.